Audio Codecs Bring Professional Sound Quality To PCs, Consumer Products

Emerging applications are driving audio codecs to 24-bit resolution, a 192-kHz sampling rate, and a dynamic range approaching over 120 dB.

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Art courtesy of AKM Semiconductor

The insatiable quest for better sound quality in low-cost PCs and digital consumer gadgets, including the emerging Internet appliances, is forcing designers to extend professional-quality audio signal processing to the mass market. As these price-sensitive products leap forward in audio performance, the high-end professional systems aren't sitting still. Top-of-the-line data converters and clever digital-signal-processing (DSP) techniques continue to push the professional audio envelope to yet a higher level for the hard-core audiophiles. Nevertheless, this performance gap continues to narrow as high-performance audio codecs with integrated bells and whistles come down in cost and power consumption.

Higher resolution, wider dynamic range, and faster update rates are the norm in the professional digital-audio world. But with refinements in price and performance of audio codecs that integrate analog-to-digital converters (ADCs) and digital-to-analog converters (DACs) on the same chip, these capabilities are now being extended to the PC and consumer worlds. Therefore, new levels of performance are reaching mainstream audio applications. As a result, the high-end segment of this audio niche is being targeted by codecs approaching 24-bit resolutions and 96-kHz sampling rates, with better than 100-dB dynamic ranges. In fact, some emerging consumer applications like DVD audio are further driving these devices to 192-kHz sampling with a dynamic range approaching 120 dB and better.

For price-sensitive PC applications, however, where 16- to 20-bit resolution is adequate, solutions that comply with the Audio Codec Standard (AC '97) are striving to strike the right chord between high fidelity and cost. The earlier versions of Intel's AC '97 specifications were simply aimed at recording and playing back from a PC. Now, the latest revision, 2.1, takes this capability to the Internet. It details techniques for linking digitized audio to the modem port, or for playing back digitized audio packets received by the modem over the Internet. In fact, prompted by standards like MP3, the playback of audio clips downloaded from the Internet is turning into a huge market that's projected to explode in the

next couple of years. Presently, the lack of security and shortage of flash memory cards have curtailed its growth.

But Forward Concepts, a market research firm based in Tempe, Ariz., predicts that the problems that have plagued portable Internet audio players will be resolved this year. Consequently, the market for these gadgets will quadruple this year to four million units, with the potential to surge to 30 million units by 2002, according to Forward Concepts' report on Internet Audio-Beyond MP3.

Meanwhile, development efforts are in progress to deliver audio codecs that adhere to multiple standards, including handling multiple channels in compliance with standards like MPEG-4 and digital theater sound (DTS). Plus, these parts provide seamless links to DSP processors and microprocessors, along with interfaces to a variety of buses like the PCI and USB.

Some key players supporting these efforts include Cirrus Logic, Analog Devices Inc., AKM Semiconductor, Burr-Brown Corp., ESS Technology, National Semiconductor, SigmaTel Inc., STMicroelectronics Inc., Texas Instruments Inc., and Wolfson Microelectronics. These suppliers of audio chips are combining single and multibit delta-sigma architectures with fine-line CMOS processes to deliver integrated codecs in miniature packages that run on lower voltages and consume an insignificant amount of power. As these developers migrate toward 0.35-µm CMOS and finer geometries for current and future codecs, they are closing the gap between standalone data converters and integrated codecs.

Cirrus Logic, for instance, has accelerated the development of audio codecs as it races to push the dynamic range of these devices to 125 dB. While 24-bit resolution, multibit architectures are first being explored in standalone DACs, the supplier intends to extend these concepts to integrated codecs later. Meanwhile, it's investigating ways to extend the dynamic range of single-bit codecs to over 110 dB. Higher dynamic range lowers the audio codec's noise floor to enable more accurate sound reproduction. Multichannel solutions are in the works too. These codecs have been optimized to interface directly to the company's line of DSP processors, eliminating the need for an external buffer between the DSP and the codec.

Meanwhile, to boost the audio performance of multimedia PC systems, Cirrus Logic has released the AC '97 v2.1-compliant stereo audio codec with a Sony/Philips digital interface (S/PDIF) digital output. The CS4297A paves the way for Microsoft's PC 98- and PC 99-compliant desktop, portable, and multimedia PCs, where high-quality audio is required, asserts the manufacturer. The CS4297A implements on-chip a 20-bit stereo DAC and an 18-bit stereo ADC with 120-dB dynamic range.

While that codec was unwrapped last year, a recent introduction is aimed at the multichannel audio/video receivers incorporated in home theater systems. Supporting multiple standards, the six-channel, 96-kHz CS4228A includes stereo 24-bit ADCs and six 24-bit DACs on a single CMOS chip. The 24-bit DACs offer

a 103-dB dynamic range and signal-to-noise ratio (SNR) with total harmonic distortion plus noise (THD+N) of 90 dB. Other features of this multichannel chip include digital volume control, anti-alias and output filters, and de-emphasis filters for 32-, 44.1- and 48-kHz sampling rates. To further ease the system designers' task of completing an audio solution for a DVD player, an A/V receiver, or an automotive system, Cirrus Logic has developed chip sets that include the CS4228A codec, an IEC958 (S/PDIF) receiver like the CS8414, and a surround sound decoder like those in the CS492X and CS493xx families. The CS4228A is based on 0.35-µm CMOS process.

Concurrently, designers at Cirrus are developing higher-channel codecs, as they explore greater integration for these chips. In fact, the company is investigating the integration of DSPs with codecs to further simplify the system integrator's job. The initial goal, though, is to combine the DSP and a high-resolution DAC with PWM output on the same silicon die.

"The consumer and professional audio performance is converging as consumer products take advantage of emerging high-performance, low-cost integrated codecs," states Richard Kulavik, product manager at AKM Semiconductor, a wholly owned subsidiary of Asahi Kasei Microsystems Co. of Tokyo, Japan. "The developers of professional units are differentiating them by adding filters and high-quality amplifiers before and after the codecs used in these systems," he adds. "Nevertheless, the professional systems also are seeing a substantial cut in price, which has never been witnessed before," he continues.

Like Cirrus, AKM Semiconductor designers also are tapping the attributes of multibit delta-sigma architecture. Employing such techniques, AKM Semiconductor's designers are pushing the performance of audio codecs to 24-bit resolution with a 192-kHz output data rate and better than 110-dB dynamic range. Initially, the multibit approach is being implemented in standalone high-resolution and high-speed ADCs and DACs. But, the manufacturer intends to extend this concept to integrated codecs very soon. According to AKM, codecs derived from the new multibit schemes will be unveiled by mid-2001.

Meanwhile, for multichannel home theaters or car stereo systems, AKM Semiconductor has unveiled a 24-bit 96-kHz codec that implements two analog inputs and six analog output channels. On this chip, the ADCs employ a dual-bit scheme, and the DACs exploit the newest multibit delta-sigma architecture. Consequently, the dynamic range for the 24-bit ADC is 102 dB, while the 24-bit DACs display a dynamic performance of 106 dB. For applications that require only two channels, AKM has released a dual version with better dynamic ratings, the AK4528. For similar resolution and speed, the AK4528 furnishes slightly wider dynamic performance in comparison to the multichannel AK4527. For instance, the two-channel 24-bit ADC for the 4528 is rated at 108 dB in dynamic range compared to 102 dB for the AK4527. Similarly, the AK4528's 24-bit DAC offers 110-dB dynamic range verus 106 dB for the DAC of AK4527.

In addition, the AK4528 implements switched-capacitor filter techniques to achieve low outband noise and high jitter tolerance. An FFT curve for a 1-kHz fundamental signal sampled by the AK4528 at 48 kHz depicts the codec's loop through performance. At 106-dB SNR, the AK4528 offers a true 18 bits of resolution. "These high-resolution and high-speed parts provide an upgrade path for the users," notes Kulavik. An eight-channel version is being investigated by the supplier as well.

For portable audio applications that demand higher functionalities from a single silicon substrate, AKM Semiconductor has developed a 16-bit multi-function audio codec with an unprecedented level of on-chip integration. Included on the monolithic AK4560A die are a microphone preamplifier, an automatic level control (ALC) circuit for the microphone preamplifier, a headphone amplifier, and a speaker amplifier. Key playback features include a digital de-emphasis filter, +2-dBV lineout filter, mute, and 0 to 50-dB analog volume control. These features enable a designer to realize power-efficient, high-quality recording and playback. With a power supply range of 2.6 to 5.5 V, the AK4560A incorporates a clever multipower scheme that allows optimal power to be specified for each circuit block in the 16-bit codec. While the AK4560A is a stereo 16-bit multifunction audio codec, a four-input version also is under development. This model also includes a programmable-gain amplifier (PGA) at the ADC input.

Growth In The Consumer Market

Observing increased activity in the high-end consumer space, such as home theater systems, DVD audio, set-top boxes, and automotive audio systems, Analog Devices Inc. (ADI) has extended its reach into this entertainment sector with a leading-edge part. Combining a multibit delta-sigma architecture with 0.5-µm CMOS technology, ADI has readied a multichannel, 96-kHz, 24-bit codec that flaunts 105-dB of dynamic range and -97-dB THD+N. Additionally, it comes with many more features like six independent volume controls. They're adjustable via an SPI-compatible serial control port, gain control for ADC input, digital de-emphasis filter, and power down mode. In effect, the AD1836 offers four ADC inputs configured as two independent stereo pairs, and six DAC channels arranged as three independent stereo pairs.

While one ADC stereo pair incorporates dedicated differential inputs, the other includes a PGA and a four-input multiplexer. The PGA permits additional amplification of the input signal in 3-dB steps up to +12 dB. Although the ADC performs at an oversampling ratio of 128, the decimation filter that follows tames the output to acceptable sample rates, with a maximum of 96 kHz.

For better noise and distortion performance at the output, all of the six DACs offer fully differential output voltages. Plus, each channel comes with a built-in programmable attenuator that has a maximum of 63 dB, adjustable in 1-dB steps.

Though the AD1836 is a 5-V part, it provides separate power-supply pins for the analog and digital sections for improved audio sound. Maximum power consumption for the unit is 750 mW with a 5-V supply. It's slated for production in the fourth quarter of this year.

Presently, on the PC front, ADI has elevated the audio experience by crafting an integrated digital audio solution, the SoundMAX, for the PC motherboard. The SoundMAX solution employs the company's AD1885 codec and Windows software drivers that take advantage of the PC's CPU processing power to implement superior audio synthesis and 3D sound effects. "This innovative architecture expands the versatility in audio design applications," notes John Croteau, product line director for ADI's Integrated Audio Group.

Interestingly, the SoundMAX codec implements two novel techniques to deliver speech recognition, conferencing, and music playback that's comparable to expensive PCI sound cards. First, it uses continuous time oversampling (CTO) to maintain a 94-dB SNR at any sample rate from 7 to 48 kHz. Also, the CTO performs asynchronous rate conversion on the fly. Second, the codec employs multibit oversampling called data directed scrambling or D^2S to protect against noise and interference from other devices in the PC enclosure. "A multibit delta-sigma architecture enables better noise shaping, flexible sample rate, and higher performance in a smaller die," notes Ken Nevard, product marketing manager for digital audio products at ADI. "As a result," he adds, "the PC is equipped with

professional sound quality that's expected in business conferencing, as well as music and home theater electronics."

ADI has further augmented this solution with the development of SoundMAX 2.1, which now supports multichannel configurations. It's compatible with all AC '97 v2.1-compliant riser cards, including Intel's new communications network riser (CNR). Based on Intel's CNR specifications, the SoundMAX's multichannel extensions support up to six channels of playback for 5.1 Dolby Digital, EAX, and A3D. The CNR specs allow system vendors to offer solutions such as the SoundMAX with room and flexibility to add LAN and modem options. To bring interactive multichannel surround-sound capabilities to low-cost PCs via its SoundMAX audio solution, ADI partnered with Mountain View, Calif.-based Staccato Systems, and the U.K.'s Sensaura Ltd.

A Focus On Voice And Speech

Other suppliers competing to grab a piece of the AC '97 pie include Burr-Brown, ESS Technology, National Semiconductor, SigmaTel, Texas Instruments, and Wolfson Microelectronics. While ESS continues to tweak its parts to maintain a strong position in this volume segment, TI is a relative newcomer. Focused more on voice and speech applications, where the audio codecs are closely aligned with its DSPs, TI has finally made a move in this direction. The TLV320AIC27--an 18-bit stereo codec with two ADCs, four DACs, and a comprehensive analog mixer--is TI's AC '97 v2.1-compliant offering. To optimize performance, the digital and analog portions of the TLV320AIC27 are powered separately. In fact, the chip permits the use of 3.3 V for digital and 5 V for analog to further minimize the power consumption. Additionally, it can also operate on a single 3.3-V supply for both digital and analog sections of the codec. In line with the company's DSP analog attachment strategy, the TLV320AIC27 is streamlined to work with TI's C54x DSP family. "In doing this, we provide a total DSP solution and ease the designers tasks," says Russell Jordan, TI's strategic marketing manager for data converters.

As for the emerging voice-band applications, TI has developed 16-bit linear codecs featuring a 22-ksample/s sample rate, serial interface, extensive applications support, and 25% lower power consumption than comparable devices. The TLV320AIC10/11, boasting an 84-dB SNR for both ADCs and DACs, operate from 3 to 5.5 V and dissipate only 39 mW at an 8-ksample/s data rate. Another salient feature is the host port interface for on-the-fly reconfiguration, which permits codec registers to be reprogrammed without interrupting the DMA block transfers. Consequently, up to eight codecs can be connected to a single DSP serial port via the master/slave mode. "Many codecs available today actually hinder DSP performance with excessive interrupts and limited serial interface flexibility," says Tom Lahutsky, new product development manager for TI's data converters.

Though identical in performance, the TLV320AIC11 is designed for digital-I/O operation at 1.2 V and 1.8 V. It directly interfaces to TI's 1.8-V UC5409 and 1.2-V UVC5402 DSPs for use in low-power systems like digital hearing aids. To facilitate development in the Internet audio arena, TI has forged a collaborative deal with the U.K.'s Wolfson Microelectronics. In cooperation with TI, Wolfson will develop a new family of media delivery devices for the evolving Internet audio appliances, with full compliance to the secure data management interface (SDMI) for copyright protection.

To further boost its presence in the analog and data-converter world, TI is in the process of acquiring Burr-Brown, a major supplier of low-noise audio codecs. Besides enabling TI to move quickly to the 24-bit precision front, the acquisition will also allow the company to exploit Burr-Brown's strong position in the audio domain, thereby driving TI DSPs into audio systems. With their combined

strengths, the potential for winning new sockets in the audio arena is greatly enhanced.

National Semiconductor's AC '97 v2.1-compliant LM4549 comes with one key enhancement. It includes a sample-rate converter to reduce the processing load of the CPU in a PC application. Besides being able to convert any rate between 4 kHz and 48 kHz with a resolution of 1 Hz, the LM4549 offers 3D sound stereo enhancement. Like other key players, National Semiconductor also is pursuing the path toward 0.35-µm and finer CMOS processes for its future solutions. In reality, the developer has internally demonstrated a 3.3-V codec with a 1-V rms output.

SigmaTel, meanwhile, has expanded the choices of AC '97 codecs with the introduction of two new chips, the STAC9783/84. Each device contains the core functionality outlined in AC '97 v2.1, while eliminating several of the seldom used inputs and other such features. In short, the STAC9783/84 answers a driving need to host a complete audio subsystem at a lower cost on one chip, asserts Alan Hansford, vice president of marketing at SigmaTel. "Both the codecs come in a 28-pin SSOP package and cost less than a dollar," adds Hansford.

Unlike others, STMicroelectronics has taken the system-on-a-chip (SoC) approach. Instead of supplying standalone codecs, the maker is readying audio solutions with an on-chip codec, a DSP processor, interfaces, memory, and other necessary features. One such solution, the STA300, includes an on-chip AC '97-compliant codec. In conjunction with a quad power half-bridge amplifier, it provides a total solution for multistandard audio applications. Armed with DSP and mixed-signal capabilities, STMicroelectronics is extending its reach into the Internet appliances sector. Toward that goal, it's developing a 20-bit codec with an integrated DSP for MP3 systems. This device is planned for release later this year or early next year.

The major thrust at STMicroelectronics is bringing on-board the output power amplifier, and keeping it cool by going to lower voltages. But the challenge is to maintain the signal quality and dynamic range as the designs migrate toward deep-submicron geometries.